## In the Claims:

**Claim 1 (currently amended):** A method of suppressing noise in a signal, said method comprising the steps of:

estimating a signal to noise ratio for said signal;

classifying said signal to a classification;

calculating a gain for said signal using said signal to noise ratio and said classification; and

modifying said signal using said gain;

wherein said calculating step calculates said gain based on $\gamma_{db} = \mu_g (\sigma''_q - \sigma_{th}) + \gamma_n$, wherein $\mu_g$ is adjusted according to said classification, and wherein $\gamma_{db}$ is a gain in a db domain, $\mu_g$ is a gain slope, $\sigma''_q$ is a modified signal-to-noise ratio, $\sigma_{th}$ is a threshold level, and $\gamma_n$ is an overall gain factor.

**Claim 2 (original):** The method of claim 1 further comprising a step of estimating a pitch correlation for said signal, wherein said calculating step further uses said pitch correlation.

**Claim 3 (original):** The method of claim 1, wherein said signal is one channel of a plurality of channels of a speech signal.

**Claim 4 (cancelled)**

**Claim 5 (currently amended):** The method of claim 2, ~~wherein said calculating step calculates said gain based on~~ $\cancel{\gamma_{db} = \mu_g (\sigma''_q - \sigma_{th}) + \gamma_n}$~~;~~ wherein $\mu_g$ is <u>further</u> adjusted according to ~~said classification and~~ said pitch correlation~~, and~~

~~wherein~~ $\cancel{\gamma_{db}}$ ~~is a gain in a db domain,~~

$\cancel{\mu_g}$ ~~is a gain slope,~~

$\cancel{\sigma''_q}$ ~~is a modified signal-to-noise ratio ("SNR"),~~

$\cancel{\sigma_{th}}$ ~~is a threshold level, and~~

$\cancel{\gamma_n}$ ~~is an overall gain factor~~.

**Claim 6 (original):** The method of claim 1, wherein said signal is in a time domain, and said method further comprises a step of converting said signal from said time domain to a frequency time prior to said estimating step.

**Claim 7 (previously presented):** The method of claim 1, wherein said signal is in a frequency domain, and said method further comprising a step of converting said signal from said frequency domain to a time domain after said modifying step.

**Claim 8 (currently amended):** A method of suppressing noise in a signal having a first signal portion and a second signal portion, wherein said first signal portion is a look-ahead signal of said second signal portion, said method comprising the steps of:

computing a voicing parameter using said first signal portion;

estimating a signal to noise ratio for said second signal portion;

calculating a gain for said second signal portion using said signal to noise ratio and said voicing parameter; and

modifying said signal using said gain;

wherein said calculating step calculates said gain based on $\gamma_{db} = \mu_g (\sigma''_q - \sigma_{th}) + \gamma_n$, wherein $\mu_g$ is adjusted according to said voicing parameter, and wherein $\gamma_{db}$ is a gain in a db domain, $\mu_g$ is a gain slope, $\sigma''_q$ is a modified signal-to-noise ratio, $\sigma_{th}$ is a threshold level, and $\gamma_n$ is an overall gain factor.

**Claim 9 (canceled)**

**Claim 10 (original):** The method of claim 8, wherein said voicing parameter is computed by a speech coder.

**Claim 11 (previously presented):** The method of claim 8, wherein said voicing parameter is a speech classification of said first signal portion.

**Claim 12 (previously presented):** The method of claim 8, wherein said voicing parameter is a pitch correlation of said first signal portion.

**Claims 13-14 (cancelled)**

**Claim 15 (original):** The method of claim 8, wherein said signal is in a time domain, and said method further comprises a step of converting said signal from said time domain to a frequency time prior to said estimating step.

**Claim 16 (previously presented):** The method of claim 8, wherein said signal is in a frequency domain, and said method further comprising a step of converting said signal from said frequency domain to a time domain after said modifying step.

**Claim 17 (currently amended):** A noise suppression system comprising:

a signal to noise ratio estimator;

a signal classifier;

a signal gain calculator; and

a signal modifier;

wherein said estimator estimates a signal to noise ratio of said signal, said signal is given a classification using said signal classifier, said signal gain is calculated based on

said signal to noise ratio and said classification using said calculator, and wherein said signal modifier modifies said signal by applying said gain; and

wherein said calculator calculates said gain based on $\gamma_{db} = \mu_g (\sigma''_q - \sigma_{th}) + \gamma_n$, wherein $\mu_g$ is adjusted according to said classification, and wherein $\gamma_{db}$ is a gain in a db domain, $\mu_g$ is a gain slope, $\sigma''_q$ is a modified signal-to-noise ratio, $\sigma_{th}$ is a threshold level, and $\gamma_n$ is an overall gain factor.

**Claim 18 (original):** The system of claim 17 further comprising a signal pitch estimator for estimating a pitch correlation of said signal for use by said gain calculator.

**Claim 19 (original):** The system of claim 17 further comprising a frequency-to-time converter to convert said signal from a frequency domain to a time domain.

**Claim 20 (currently amended):** A system capable of suppressing noise in a signal having a first signal portion and a second signal portion, wherein said first signal portion is a look-ahead signal of said second signal portion, said system comprising:

a signal processing module for computing a voicing parameter of said first signal portion;

a signal to noise ratio estimator;

a signal gain calculator; and

a signal modifier;

wherein said estimator estimates a signal to noise ratio of said second signal portion, said second signal portion gain is calculated based on said signal to noise ratio and said voicing parameter using said calculator, and wherein said signal modifier modifies said second signal portion by applying said gain; and

wherein said signal gain calculator determines said gain based on $\gamma_{db} = \mu_g (\sigma''_q - \sigma_{th}) + \gamma_n$, wherein $\mu_g$ is adjusted according to said voicing parameter, and wherein $\gamma_{db}$ is a gain in a db domain, $\mu_g$ is a gain slope, $\sigma''_q$ is a modified signal-to-noise ratio, $\sigma_{th}$ is a threshold level, and $\gamma_n$ is an overall gain factor.

**Claim 21 (canceled)**

**Claim 22 (original):** The system of claim 20, wherein said signal processing module is a speech coder.

**Claim 23 (previously presented):** The system of claim 20, wherein said voicing parameter is a speech classification of said first signal portion.

**Claim 24 (previously presented):** The system of claim 20, wherein said voicing parameter is a pitch correlation of said first signal portion.

**Claim 25 (cancelled)**

**Claim 26 (previously presented):** The system of claim 20 further comprising a frequency-to-time converter to convert said second signal portion of said signal from a frequency domain to a time domain.